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LEVEL ADJUSTMENT OF AUDIO SIGNALS BY MEANS OF A SOLID-STATE ELECTRONIC ATTENUATOR

BY

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LEVEL ADJUSTMENT OF AUDIO SIGNALS

BY MEANS OF

A SOLID-STATE ELECTRONIC ATTENUATOR

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R. BECK

McCURDY RADIO INDUSTRIES LIMITED

Last year, our company was approached by the Westinghouse Broadcast Company to build a completely solid-state, centralized, radio broadcast center for Radio Station WINS in New York. The requirement dictated that all switching, selection, and level adjustment must be achieved remotely, as no audio signal paths were to exist in the audio control desks.

Switching and selection were to be done by means of highly reliable relay switching techniques. Solid-state switching, using diodes and transistors, was examined and it was concluded to be economically unsound for this application. Relays, with present day switching techniques, proved more than sufficiently reliable for this particular application of an audio switching and selecting system.

The next problem to be considered was how to adjust and vary the level of various audio signals with the same degree of precision and accuracy as already achieved with present-day mechanical stepped attenuators used by the broadcast industry.

The method that first comes to mind, and which has been in use for the past few years with fair success for adjustment of audio levels, was the Light Dependent Resistor. This incorporates a cadmium-sulfide cell whose resistance is proportional to the amount of light falling on it. This means that an electric lamp with a filament must be used, and was ruled out as not being a solid-state device. In addition, the response time to changes in level was found to be too slow for many applications of audio mixing; further, these devices exhibit a hysteresis effect which, of course, would not permit the same amount of attenuation repeatedly for the same

setting of the attenuator control.

Also considered were rf modulation and pulse gate techniques, field-effect transistors, and also some types of variable-mu transistors, which have been used for automatic gain control in rf circuitry. These devices presented so many insurmountable problems, such as cost, dynamic range, distortion, amount of attenuation, and dc bias shifts which develop undesirable low-frequency transients in the output, to name just a few, that they were dismissed.

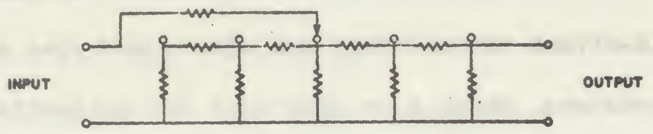
The method of attenuation finally settled upon was that of using the variable-impedance characteristic of a forward-biased diode. The diode impedance decreases proportionately to the forward dc current flowing through it. Of course, this method has its defects, particularly variations with temperature.

Before going on to discuss this method of attenuation, let us examine in some detail the requirements to be met by the solid-state electronic attenuator.

ATTENUATOR REQUIREMENTS

To see what the attenuator requirements should be, a study was made of the existing mechanical mixer-attenuators presently being used by the broadcast and recording industries.

They are basically a variable resistive network with attenuation achieved by switching in successive steps from one resistive value to another (see Fig. 1). Their deterioration to frequency response and introduction of distortion over the audio band of frequencies from 20 cps to 20 kc is almost non-existent.



BASIC CONFIGURATION FOR AN UNBALANCED

LADDER ATTENUATOR NETWORK

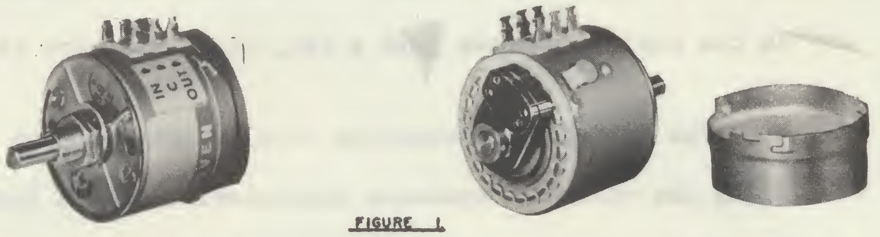
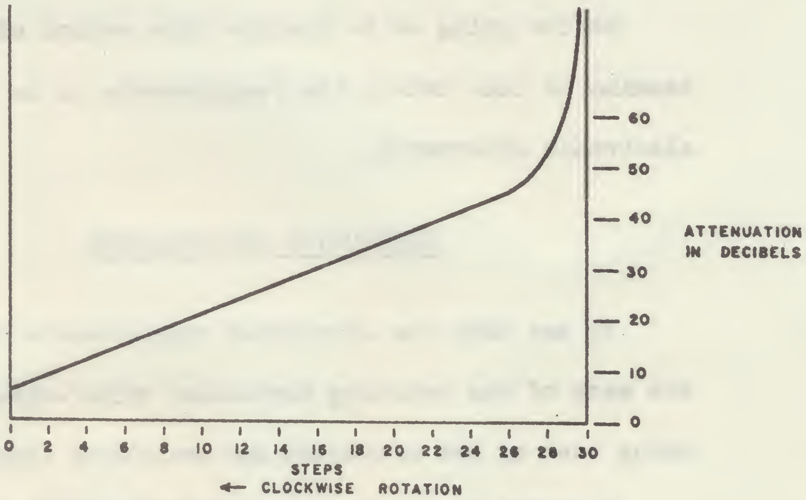
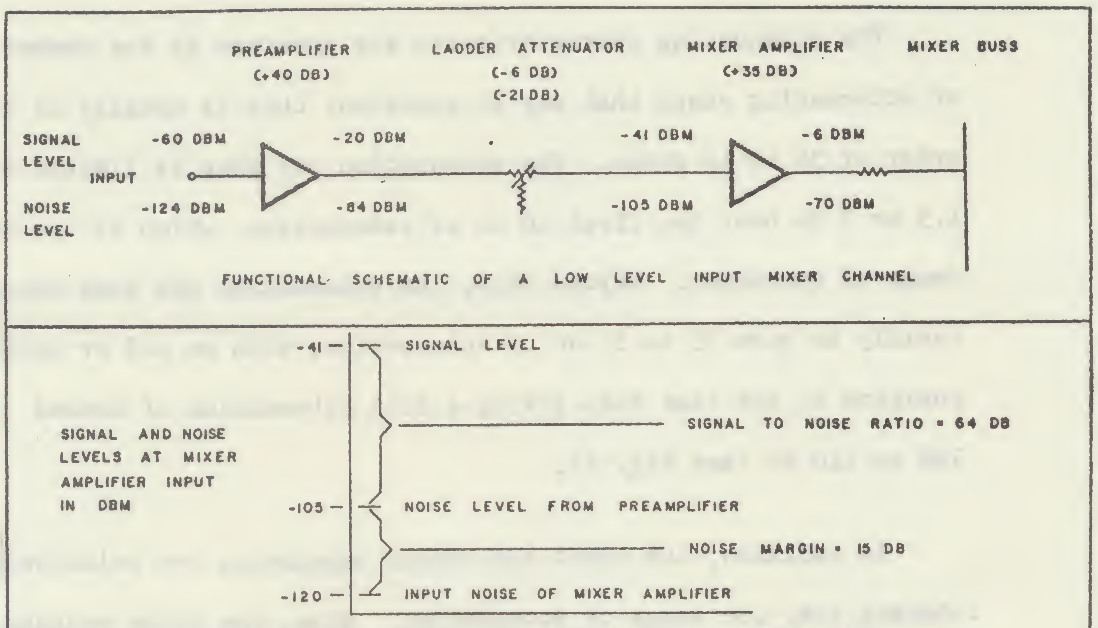


FIGURE 1.



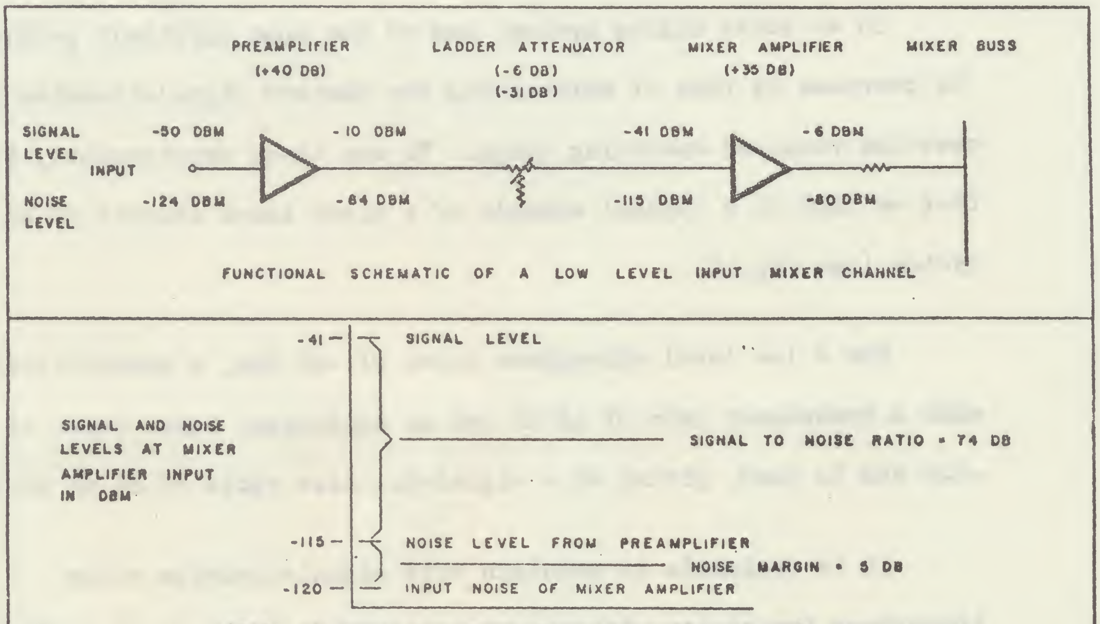
TYPICAL ATTENUATING CHARACTERISTIC OF A STEPPED
LADDER ATTENUATING NETWORK

FIGURE 2.



GRAPH OF SIGNAL TO NOISE RATIO AT MIXER AMPLIFIER INPUT
(ATTENUATOR SET AT 15 DB "IN HAND")

FIGURE 3



GRAPH OF SIGNAL TO NOISE RATIO AT MIXER AMPLIFIER INPUT
(ATTENUATOR SET AT 25 DB "IN HAND")

FIGURE 4

If a ladder type of attenuator is used which has a 6 db inherent loss, the total loss for the signal and noise would be 21 db; so that at the input to the mixer amplifier the signal is at a level of -41 dbm and the equivalent input noise from the microphone preamplifier is -105 dbm.

The equivalent input noise of the mixer amplifier is -120 dbm and, as a result, there is 15 db of margin between the two noise levels; that of the microphone preamplifier predominates, and the signal-to-noise ratio is still 64 db.

The gain of the mixer amplifier is 35 db (matched), giving a signal level to the mixer bus network of -6 dbm and an equivalent noise level of -70 dbm.

These are levels for normal program settings, but to give our audio system a greater operating dynamic range, the input level at the microphone preamplifier must be increased another 10 db to -50 dbm. This provides a signal-to-noise ratio of 74 db at the input, which is to be maintained throughout the system (see Fig. 4).

Under this condition, the level into the mixer attenuator is now -10 dbm, but the equivalent noise is still -84 dbm. Also, the output level to the mixer bus must be kept constant; therefore, the attenuation on the mixer attenuator must be increased by an additional 10 db, to 25 db in hand.

The levels into the mixer amplifier are now -41 dbm for the signal (same as previous) but the input equivalent noise from the preamplifier has dropped 10 db to a level of -115 dbm.

As can be seen, there is only a difference of 5 db between the input noise of the mixer amplifier and that contributed by the preamplifier. This is the lowest permissible limit before the input noise of the mixer amplifier begins to become effective and the signal-to-noise ratio set at the microphone preamplifier input will begin to deteriorate.

The last thing to consider is the maximum input level that the attenuator must handle. The microphone preamplifier has a transducer gain of 40 db and can handle a maximum input level of -20 dbm; therefore, the maximum level the attenuator must be capable of handling without distortion is +20 dbm.

Thus, the requirements we set down for the electronic attenuator are as follows:

1. It is to replace the mechanical attenuator and the mixer amplifier as one complete unit, and provide the mixer amplifier's gain of 35 db.
2. Have an input impedance of 150 or 600 ohms balanced or unbalanced, and handle a maximum input level of +20 dbm.
3. Deliver a maximum output level of +10 dbm into a 150 or 600-ohm load, balanced or unbalanced.
4. The frequency response from 20 cps to 20 kc must be within 0.5 db, referred to 1 kc.
5. The total harmonic distortion at maximum levels must be within 0.5%, 30 cps to 20 kc; at operating levels, within 0.25%, 30 cps to 20 kc. The intermodulation distortion at maximum levels must be less than 1%; at operating levels less than 0.5%.
6. To exhibit a modified logarithmic attenuating curve over a range of 85 db and have an infinite or off position of -120 db or greater.

7. Should be able to achieve the same signal-to-noise and dynamic range characteristics previously discussed for audio mixing channels.
8. Have a stability of ±1 db, 0°C to 55°C.
9. Exhibit an equivalent attenuator noise of -85 dbm or better.
10. Operate off a 36 v dc regulated supply.
11. Six units to fit in a standard 1-3/4" x 19" rack space.

THE OPERATION OF THE RG243 REMOTE GAIN AMPLIFIER

General Description

The RG243 is an electronic attenuator and amplifier designed to become an integral part of an audio mixing system (see Fig. 5). The unit is designed to occupy only 1-3/4 inches of vertical space in a standard 19-inch broadcast rack, with six units fitting across the width of the rack (see Fig. 6). The dimensions of the individual unit are 1-3/4 inches high, 2-7/8 inches wide, and 9-1/2 inches long; it weighs approximately 2-1/2 lb.

The method chosen to achieve attenuation electronically, and by means of solid-state devices, was that of changing the forward-conducting impedance of silicon diodes. This is not a new idea and has been in use by Bell Telephone for many years. Also, several manufacturers to date are employing this principle in peak-limiting amplifiers. The term Vario-Losser is often used to describe this method of electronic attenuation.

The RG243 may be broken into three separate sections (see Fig. 7):

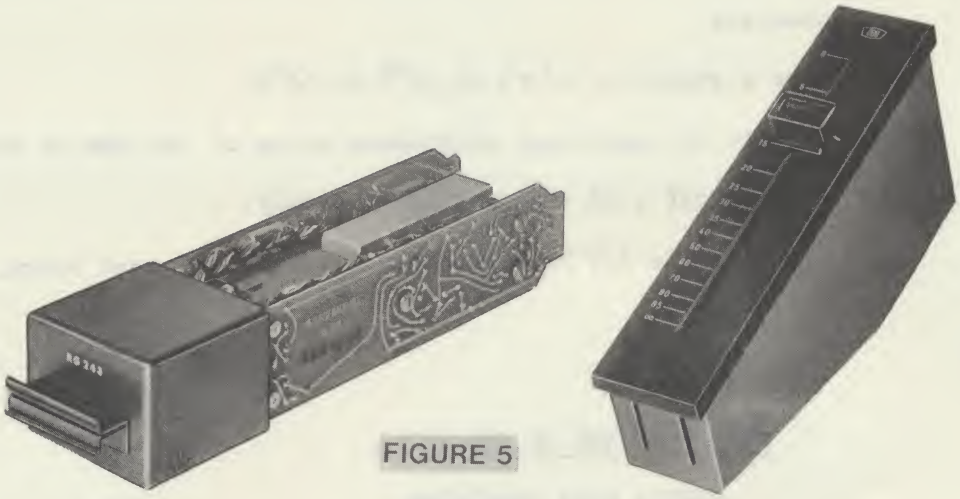


FIGURE 5

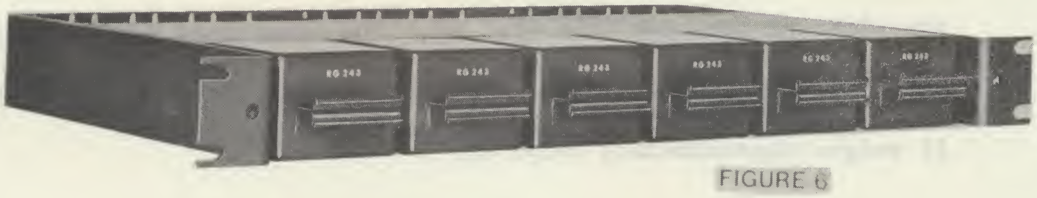


FIGURE 6

(a) The electronic attenuator.

(b) The dc control amplifier which drives the electronic attenuator.

(c) An audio amplifier which follows the attenuator.

The attenuator handles a maximum input of +20 dbm (8v rms approx.) at an input impedance of 150 or 600 ohms, balanced or unbalanced. The nominal input is -20 dbm (0.077v rms across 600 ohms).

The electronic attenuator may be subdivided into two parts (see Fig. 8):

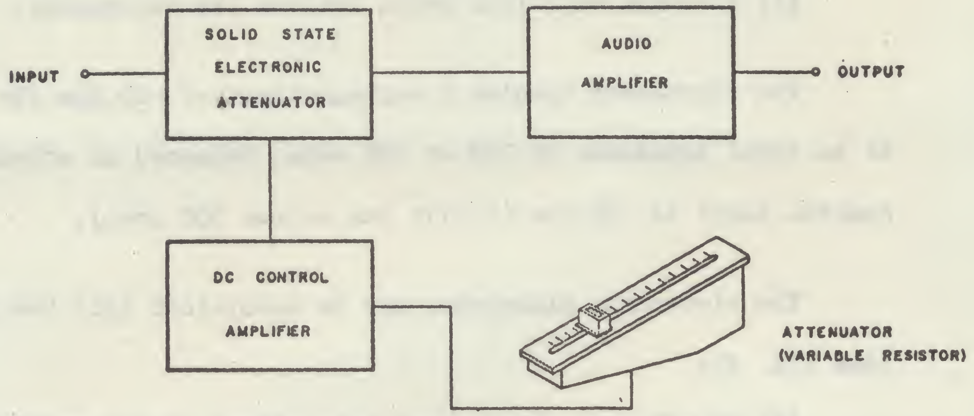
(a) An off-gate which comes into effect at the maximum attenuation position, giving an attenuation of -125 db or greater.

(b) The attenuator network.

<u>Attenuator Network</u>

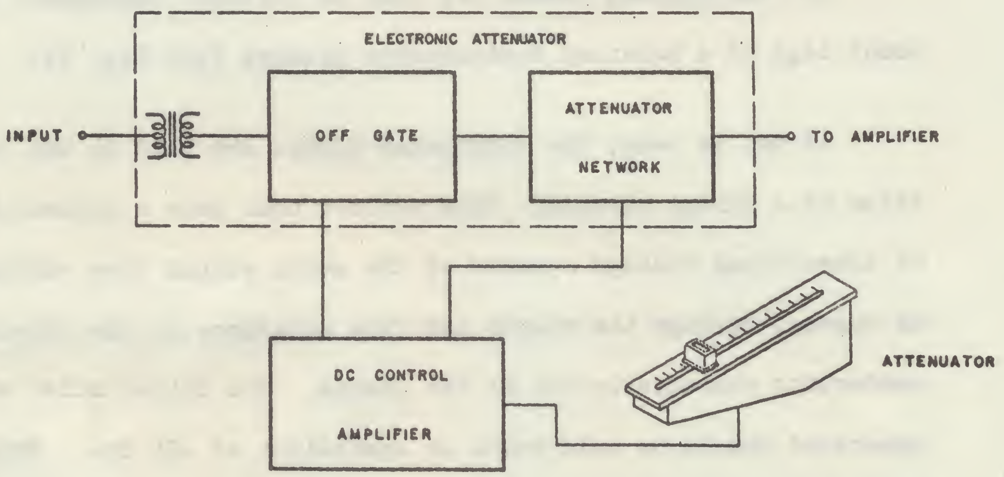
The attenuating diodes are used as variable impedances in the shunt legs of a balanced H-attenuator network (see Fig. 9).

As can be seen, the attenuator diodes are used in the configuration of a bridge circuit. This ensures that only a minimal amount of interfering voltage appears at the audio output from varying the dc current through the diodes and from unbalance in the forward-conducting characteristics of the diodes. The output noise voltage generated should be held below an equivalent of -85 dbm. This dictates that the diode forward-conduction characteristics must be matched to a very close tolerance over the current range employed (0 to about 40 ma). This tolerance is approximately 1 ua. Originally, this appeared to be almost an overwhelming complication but was gradually overcome and now is achieved with relative ease. The distortion introduced by this attenuator network is in the order of 0.1% to 0.15%.



BLOCK DIAGRAM OF REMOTE GAIN AMPLIFIER

FIGURE 7



BLOCK DETAIL OF ELECTRONIC ATTENUATOR

FIGURE 8

The attenuator network gives up to 85 db or more attenuation, after which the diode off-gate conducts and provides an additional shunt leg to the attenuator which increases the attenuation to -125 db or greater from 20 cps to 20 kc.

DC Control Amplifier

The dc control amplifier driving the electronic attenuator may be separated into three parts (see Fig. 10):

(a) Attenuator drive amplifier.

(b) Attenuator off-gate control.

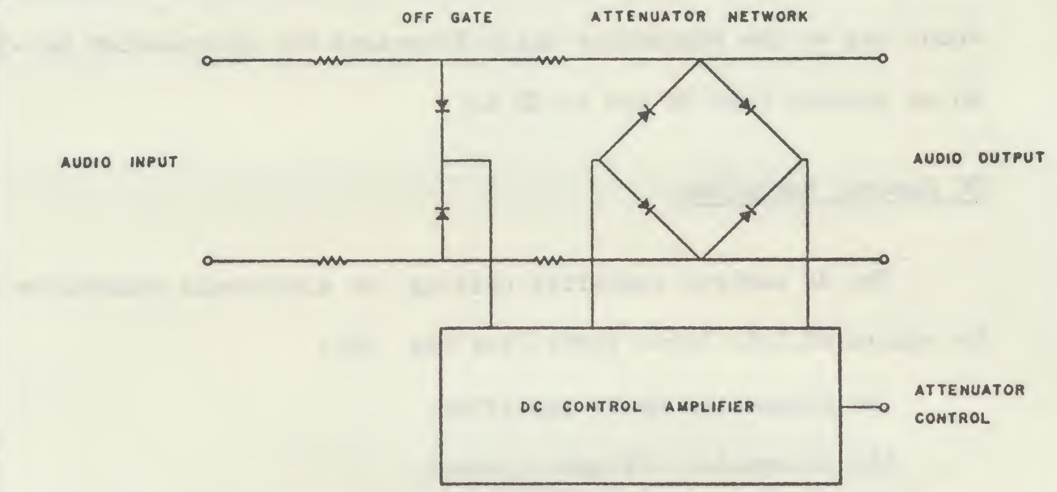
(c) Power supply decoupler.

The attenuator drive amplifier does several important functions besides varying the amount of dc current through the attenuator network.

(a) One of the biggest drawbacks or weaknesses of the diodes is their temperature sensitivity. The forward voltage drop across a diode will drop at a rate of approximately 2.5 mv/C^{o} which must be compensated for. Also, the variation in current through the diode will change its temperature and thus change the amount of attenuation it represents.

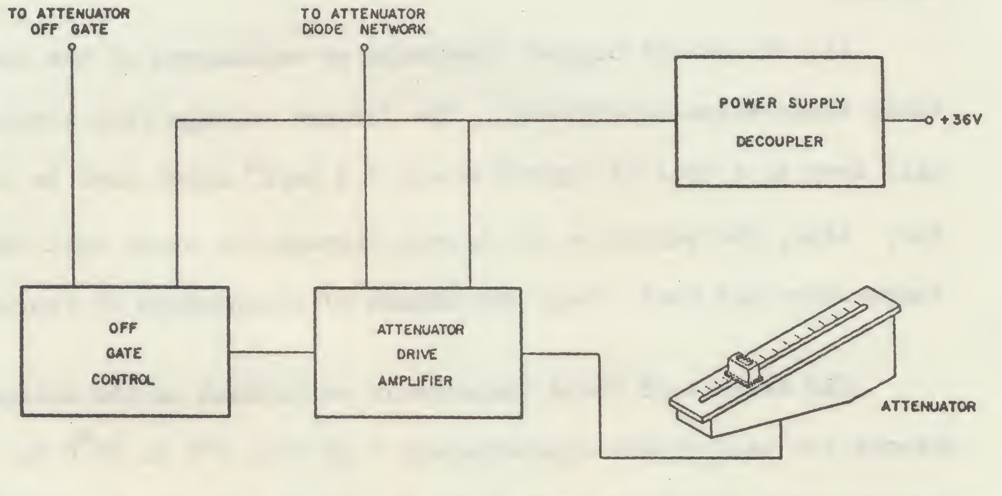
The effects of these temperature variations on the attenuator network are held within approximately 1 db from 0^{o}C to 55^{o}C by encapsulating within an epoxy block the diode attenuator network, and a section of the attenuator drive amplifier with its temperature compensating components.

(b) It determines the taper or attenuating characteristics as a function of the rotational position of the external variable resistor. It has been set up to provide a modified logarithmic taper (see Fig.11).



SCHEMATIC OF ELECTRONIC ATTENUATOR

FIGURE 9



BLOCK DETAIL OF D.C. CONTROL AMPLIFIER

FIGURE 10

The external variable resistor is a precision 5K linear control and requires only two wires. Only a few microamps of current flows; therefore, wire length is of little consequence and is easily compensated for by a "gain set" control in the attenuator drive amplifier. The attack time, or response of the attenuator network to control-current change, is in the order of microseconds but is slowed down to several milliseconds by a capacitive time constant network to give a smoother attenuating characteristic.

(c) It sets the point of attenuation where the cut-off gate circuit is operated to give full attenuation.

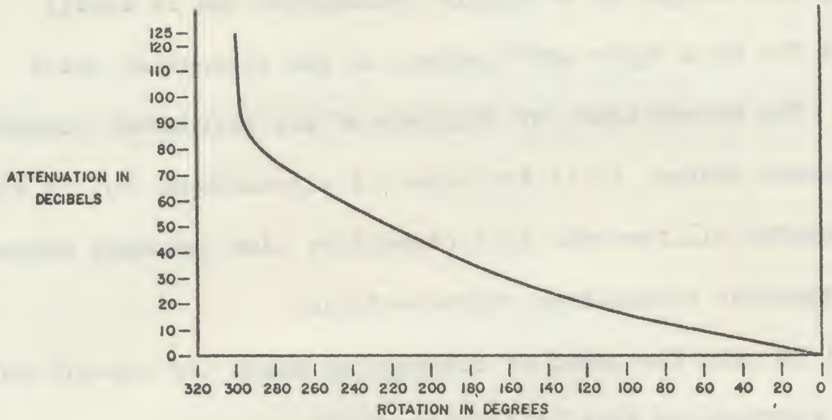
The attenuator drive amplifier and off-gate are isolated from the 36v dc regulated supply by a capacitive multiplier type of series regulator. This is to prevent any possible supply transients or other attenuators from causing interaction when operated off the same supply.

This electronic attenuator gives more attenuation at cut-off than most mechanical step-attenuators, as well as giving a continuous attenuating characteristic down to 85 db which permits use of the full attenuating range for mixing purposes.

AUDIO AMPLIFIER

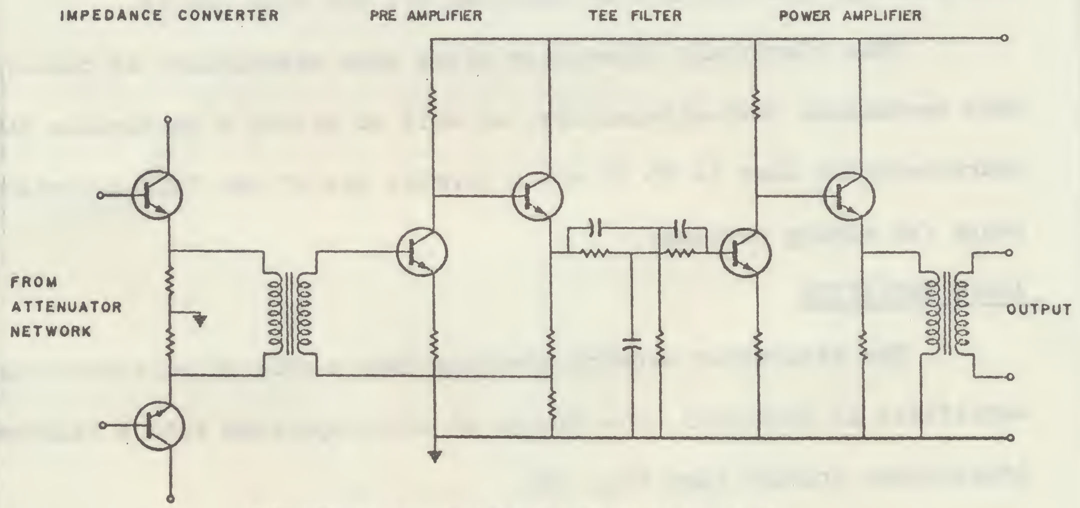
The attenuator network operates into a pair of emitter-follower amplifiers in push-pull, the output of which operates into a balanced transformer primary (see Fig. 12).

The secondary of the transformer is an integral part of a low-noise, high-gain preamplifier stage. The low noise of this stage sets the limit of how much attenuation may occur before the signal-to-noise ratio at the attenuator input is degraded. Normal operation of the unit in a system will permit as much as 25 db attenuation before the signal-to-noise ratio is degraded.



ATTENUATION CURVE OF ELECTRONIC ATTENUATOR

FIGURE 11



BASIC SCHEMATIC OF ATTENUATOR OUTPUT CIRCUIT AND AUDIO AMPLIFIER

FIGURE 12

The low-noise preamplifier operates into a parallel-Tee notch filter which limits sharply low-frequency components below 20 cps (the attenuation is better than 15 db at 10 cps and 20 db at 5 cps). The filter network serves the purpose of removing low-frequency transient components introduced by the slight unbalance and by the dc control current in the diode attenuator network.

The parallel-Tee filter output operates into a common-emitter amplifier which brings the total amplifier gain to 35 db.

The common-emitter amplifier drives an emitter follower which provides the required power gain and low output impedance. The emitter follower couples to an output transformer which will deliver a maximum level of +10 dbm into 600 ohms or 150 ohms, balanced or unbalanced. The nominal output level is 0 dbm.

The total harmonic distortion introduced by the entire attenuator and amplifier is better than 0.5% 30 cps to 20 kc at a maximum output of +10 dbm, and less than 0.25% 30 cps to 20 kc at normal operating levels. Frequency response is within 0.5 db from 20 cps to 20 kc.

This philosophy of remote control of audio signal levels is another step towards realization of the broadcast engineers' desire of having a broadcast system with all electronic equipment centralized in one location. Another application of the remote gain amplifier is public address systems, or in theatres, where the audio control desk may be completely portable and positioned in any desired operational area with simple dc remote control wires returning to the active equipment center.

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AN IMPROVED TRANSISTORIZED SOUND SYSTEM FOR STUDIO USE

BY

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98th TECHNICAL CONFERENCE

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An Improved Transistorized Sound System For Studio Use

Summary. The exclusive use of silicon-planar transistors and printed wiring boards together with constructional elements of multiple application characterize a range of new units for studio use developed under the name of SITRAL. The small dimensions and the low power consumption of the units permit compact assembly of small or large, mobile or stationary mixing and control desks for monaural or stereo operations.

The still increasing demands on the sound regarding the artistic performance and the technical possibilities in the studios for radio, television, sound film and disc recording call for an improved and more comprehensive technical outfit. Besides the units required for distributing the modulation in the studio as well as for receiving the signals from outside, the centralized sound-control equipments of the studios, i.e. the mixing and control desks to be realized by the desired compact construction, are concerned in the first place. In the main, these centralized sound-control equipments must meet the following exacting requirements:

Increased number of input channels;
Possibility to form several groups, including the facility to allot each channel to any of these groups;
Increased utilization of equalizing and amplitude limiting devices;
Wide range of control without the risk of overmodulation for the individual channels as well as for the groups;
High signal-to-noise ratios;
Applicability to stereophonic operation via two to six channels;
Higher constancy of operation;

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Neat and clear arrangement of the considerably increased number of control elements;
Small dimensions and small power consumption, permitting compact constructions.

All these requirements can be satisfied only by the use of fully transistorized units. Fortunately, recent progress in the field of semi-conductors has also led to the development of transistors, which meet the extreme demands made upon the quality of studio equipments. In the laboratories of Siemens & Halske AG a new line of equipment, the so-called SITRAL*) technique, has, therefore, been developed. The characteristic feature of construction, common to all SITRAL apparatus, is the use of printed wiring boards and silicon-planar transistors.

First of all, the characteristics of this system and the technical data obtained will be considered.

1. Characteristics of the system

1.1. Electrical features

The structure of the SITRAL system, i.e. the functions of the individual links of the chain of units may be seen from Fig. 1 representing the basic circuit diagram of a sound-studio equipment. The characteristics of the system are first of all influenced by the quality of the pre-amplifier, the equalizer, the group amplifier, and the output amplifier. Frequency response, non-linear distortion coefficient and power consumption as well as range of modulation, signal-to-noise ratio and operational reliability are decisive for the quality.

An important fact is the use of silicon-planar transistors in all stages. Owing to their favourable noise

*) trade mark
(Silizium-Transistoren-Leiterplattentechnik)

figure, their high cutoff frequency, their small leakage current, and their high admissible junction temperature, they enable the extreme demands to be met, which are made on the quality of studio equipments. The life expectancy of these transistors is longer than that of other components used in amplifiers. The SITRAL amplifiers may, therefore, be considered to be almost immune against wear and tear.

The operating voltage of all SITRAL units is 24 v d-c. The power consumption is so small that no special cooling provisions need be taken. The non-linear distortion factor (Fig. 2) remains below 0.5 % within the whole transmission range, the output impedance of the output stages being $\leqq$ 50 ohms, the load resistance 300 ohms. The frequency response within the limits of 40 and 15,000 cps shows a linear characteristic with a maximum deviation of $\pm$ 0.5 db. These data are guaranteed within the range of ambient temperatures of -20^{o} ... $+60^{o}$C.

For judging the quality of mixing and control desks, the above data as well as the noise values and the modulation limits are not sufficient; the behaviour of the whole chain of units is decisive. For instance, that system is the better one which has the greater signal-to-noise ratio under the various possible operating conditions, taking into account the maximum admissible fader attenuation in channel and groups without the risk of overmodulation. It, therefore, comes in very useful that the "Institut für Rundfunktechnik (IRT)" have elaborated a new method for determining and representing the essential operating characteristics of such equipments on the basis of standardized rules, relating among

others to number of channels, amplification, fader adjustment and microphone characteristics. Thus, it is now possible to detect, easily and unambiguously, the working qualities of the individual systems, and to compare and judge them objectively. The following statements are based on this method.

The operating characteristics show the relation between the dynamic volume range at the output and that at the input of the chain of units. The latter is determined by the type of microphones used, and by this also the input voltage corresponding to a certain dynamic value. The 45^{o}-line represents the behaviour of an ideal system. All deviations therefrom are impairments caused by the units. This representation shows clearly the influence exerted by the group and output amplifiers on the dynamic volume range, likewise the adaptability of the pre-amplifier to the various input levels.

Under otherwise equal conditions, that system will facilitate the practical work of the sound-engineer which has the wider control range utilizable without the risk of over-modulation. Fig. 3 shows the admissible fader adjustments of the SITRAL system, making allowance for an over-modulation reserve of 6 db for all units of the chain. This over-modulation margin is necessary on account of sudden level peaks. The "maximum fader adjustment" is 34/46 db. This means that the group fader can be adjusted to a maximum attenuation of 34 db without the risk of over-modulation, while the channel and group faders can be adjusted conjointly, in any combination desired, to a total attenuation of 46 db.

A performance taking into account this fader reserve

is represented by the "mean operating characteristic". The curves of this graph are related to mid-positions of the faders, from where the setting members of the latter can be displaced to either side by equal values, in the present instance by the attenuation values of 11.5 db in both the channel and the group. Due to the fact that the SITRAL system yields an over-all amplification of 110 db, the amplification during the process of plotting this characteristic is 87 db. The adaptation to the different input voltages is exclusively effected by the control devices of the pre-amplifier. Fig. 4 shows the signal-to-noise ratios obtained by using dynamic microphones, for which a sensitivity of 0.1 mv/μb and a noise level of -123 db are assumed. This and all following noise figures are based on the CCI weighting curve for measuring noise on program transmission facilities and are peak values according to German Standards. In this case, a sound pressure of 10 μb yields an output signal-to-noise ratio of 63 db, as shown by the characteristic.

When using condenser microphones having a sensitivity of 1.0 mv/μb and a noise level of -106 db, the values will be more favourable, as may be seen from Fig. 5.

1.2. Constructional features

The construction of the SITRAL units permits to meet to a great extent the manifold requirements of practical life with the aid of standardized component parts. Figs. 6a and 6b show the typical structure of a plug-in amplifier. All

components are arranged on printed circuit boards of epoxy resin. This printed wiring board has the standard dimensions of 160 mm x 100 mm. Its front is provided with a front plate and the control elements, its rear with a round-pin plug strip. The board can be inserted and fixed in a protective case forming thus a plug-in unit. Four guiding bars at the protective case assure exact guiding when sliding the plug-in unit into the holder. A perforated bar fixed on the protective case together with a locking device on the wiring frame prevent the insertion of wrong units. The coding, i.e. matching individual perforations in the bar with corresponding arrangements of the locking pins offers numerous variations.

The front plates of all channel units have a width of 40 mm matching the width of the standardized plane faders. As a result, all units of a channel can be clearly arranged in one row, a salient point for neat and clear arrangement.

The holder (Fig. 7) has the task to house the individual units and to establish the electrical connections. It consists of two robust, pressure-cast side walls, the extruded sectional bars as well as of the guide bars for the plug-in units, the wiring frame with the spring-contact strips and the afore-mentioned locking device.

Owing to the fact that not all units of the SITRAL chain have a width of 40 mm, the guide bars in the holder for the plug-in units are arranged in an especially flexible way allowing thus to choose the position of the guide bars at intervals of 2.5 mm. The holders can be made of any length desired. Hence desks and cabinets of any width can be equipped with the same unitized basic elements.

The electrical connections between the individual plug-in units and the wiring of a desk are established at the wiring frame by means of gilded plugs and spring-contact strips.

Due to their constructional features, the SITRAL units may be used in desks of any construction. The profile of a standardized design has been chosen in view of an as small as possible building depth. The arrangement shown in Fig. 8 assures good accessibility and easy supervision of the control elements, guarantees the desired small building depth and does not embarrass the operator's knees. This construction may be used with advantage as well for small, light-weight news reporting desks as for elaborate stationary desks.

2. SITRAL units

The SITRAL manufacturing program is so extensive that not all components can be described here. In the following, therefore, only those units will be considered which are essential to the quality of studio equipments.

2.1. Pre-amplifier

The pre-amplifier (Fig. 6) amplifies the voltage coming from the microphone or other sources to the level of +6 db*); the modulation limit is +21 db. The noise voltage (according to DIN 45 405) - referred to the symmetrical amplifier input (impedance >600 ohms) - is <-121 db at an amplification of 76 ... 34 db and rises slightly in case of lower amplifications. This value of -121 db obtained with low input voltages, which nearly

*) All level values mentioned here are referred to a voltage of 0.775 $\hat{=}$ 0 db.

equals the set noise of dynamic microphones (noise factor F = 2), is the basis for the extraordinarily favourable signal-to-noise ratio of the SITRAL system. A continuously adjustable potentiometer provided beside the step control, calibrated in db, permits to adjust several channels to exactly the same amplification, a fact which is very important for stereophonic recordings. A three-step switch allows to switch in at will a solid-borne-noise filter with the limit frequency of 40, 80 or 120 cps. The maximum admissible input voltage of the unit, at zero amplification, is +21 db.

2.2. Equalizers

The equalizers (Fig. 9) are equalizer-amplifiers with an amplification of 0 db. The admissible input voltage and the modulation limit are +21 db; the noise voltage, referred to the ungrounded transformer input, is $<$-95 db. Thus, the equalizers will not cause a reduction of signal-to-noise ratio even with critical operating conditions. Three equalizer types are used: i.e. a band-pass unit, a unit with treble/bass control and mid-range control as well as a unit with treble/bass control and presence filters. The last mentioned unit permits to vary the level of the limit frequencies 40 and 15.000 cps by $\pm$ 15 db, by steps of 3 db, with the aid of separate switches (Fig. 10a), as well as to select one of five presence frequencies and to raise thus the level up to 8 db by steps of 2 db (Fig.10b). The equalization characteristics of the mid-range controller are shown in Fig. 10c. Controls with fixed steps have been chosen for all equalizers in order to guarantee an unequivocal reproducibility of the adjustments determined in rehearsal.

2.3. Group amplifier

Modern sound-control equipments are characterized by the possibility of allotting or connecting the input channels to any of several groups. In the SITRAL system this problem is solved by the group amplifiers. Units with eight and with twelve inputs are available; they have an amplification of +10 ... +12 db; the modulation limit is +21 db. The noise voltage, which is especially important with these units, is $<$-100 db. If two units per group are used, the number of channels which can be allotted or connected is increased accordingly.

2.4. Limiter-compressor amplifier

The limiter-compressor amplifier ensures constant output voltage in case a certain input voltage is exceeded. It consists of signal amplifier, control amplifier and diode network. The unit can be adjusted for regulating the input voltages between 0.5 v and 1.5 v; the output voltage is 1.5 v at 300 ohms. Input and output are not grounded. The attack time is 0.5 ms and the release time is adjustable from 0.5 to 1.5 s. Remarkable features are the small non-linear distortion factor of $<$ 0.5 % for limiter operation in the frequency range 200 cps ... 15 kcps and of $<$1 % at 40 cps, as well as the signal-to-noise ratio of $\geqq$ 70 db. The control characteristics may be seen from Fig. 11.

2.5. Volume indicator

The control of modulation is performed by a mirror or a pointer instrument working together with a logarithmic amplifier. This amplifier is decisive for the course of the attack time and the release time; its transformer input is not grounded. Both indicating instruments can be

connected simultaneously. The scales are calibrated in db; the pointer instrument has, in addition, a percent scale (0 db $\triangleq$ 1.55 v $\triangleq$ 100 %). The easily readable measuring range extends from -50 ... +5 db. The attack time is 10 ms (overshot less than 1 db), and the release time is approx. 1 second.

2.6. Further units

Besides the units shown in Fig. 1 which are not described in detail on the preceding pages, the new unit chain comprises the following units:

Isolating amplifiers with a return transmission attenuation of more than 110 db.

Line amplifiers for adapting the signal taken over from the telephone line to the studio level with a return transmission attenuation of $\geqq$ 115 db and an adjustable amplification from +6 to +25 db.

Mixing amplifiers with two separately controllable inputs and an ungrounded output; cross-talk attenuation of inputs $>$ 60 db; maximum amplification 24 db.

Distributing amplifiers with separately controllable inputs having a cross-talk attenuation of $>$ 60 db, and with four separately controllable outputs; maximum amplification 9 db.

Sound direction controls and sound direction mixers as well as special equalizers for stereophonic tasks.

3. Sound-control equipments in SITRAL technique

Since the afore-described units are smaller and lighter

and consume considerably less current than valve-type units of equal performance, it is now possible to combine them to very compact sound-control equipments of any extension. A number of desk-types has been standardized and can be used with the same good results as well in film, television and radio studios as for transmission purposes.

Fig. 12 shows a standardized desk of medium size with 8 inputs, two of which are high-level inputs. It is possible to form 2 groups with 2 outputs. All channels can be allotted to any of these two groups. The power consumption of the fully equipped desk is 32 watts, which may be derived either from a battery or from a regulated power supply, since the individual units are efficiently decoupled. The desk (without carrying case) has a weight of 32 kg and can be easily transported.

Fig. 13 shows a bigger desk which is equipped with 12 low-level input channels and 8 equalizers. The 12 channels can optionally be distributed to 2 groups. Beside the equalizers the following instruments and units are arranged from left to right: the audio-frequency generator with 5 measuring frequencies, the volume indicator meter as well as the pre-listening loudspeaker with amplifier and the microphone for the command system. All operating elements can easily be reached by the sound engineer while sitting on his chair, and the desk offers space enough to place textbooks and so forth. This desk has a power consumption of 60 watts.

Texts for the Figures

Fig. 1 — SITRAL sound-control desk, simplified block diagram

Fig. 2 — Typical distortion curve of the SITRAL amplifiers

Fig. 3 — Level diagram in order to determine the "mid-position" of the faders

Fig. 4 — ~~Dynamic~~ characteristics ~~for fader mid-positions~~ Mean operating with the use of dynamic microphones

Fig. 5 — ~~Dynamic~~ characteristics ~~for fader mid-positions~~ Mean operating with the use of condensor microphones

Fig. 6a — SITRAL pre-amplifier, front and components side

Fig. 6b — SITRAL pre-amplifier, rear and circuit side

Fig. 7 — Case for SITRAL units

Fig. 8 — SITRAL sound-control desk, cross-section of the standard model

Fig. 9 — SITRAL equalizer with bass/treble control and presence filter

Fig. 10a — Treble/bass equalizer, frequency response

Fig. 10b — Presence filter, frequency response

Fig. 10c — Mid-range controller, frequency response

Fig. 11 — Limiter/Compressor amplifier, output level versus input level

Fig. 12 — 6+2/2-channel sound control desk in SITRAL technique

Fig. 13 — 12/2-channel sound control desk in SITRAL technique

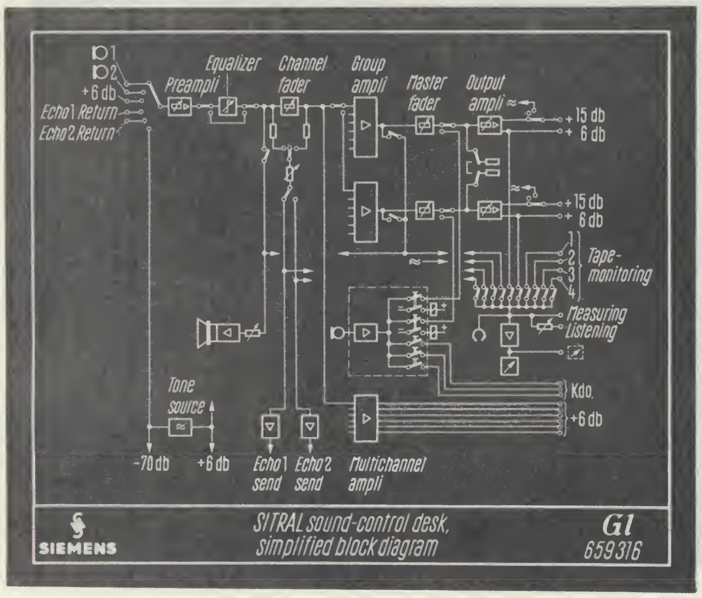


Fig. 1

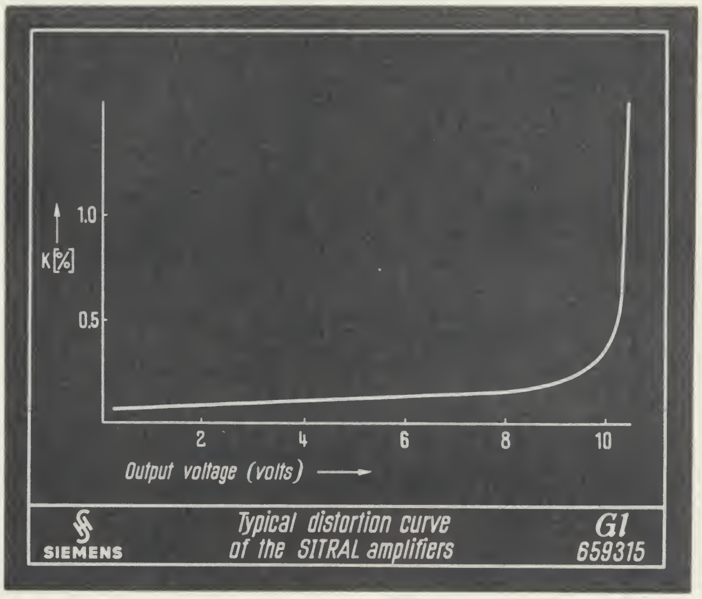


Fig. 2

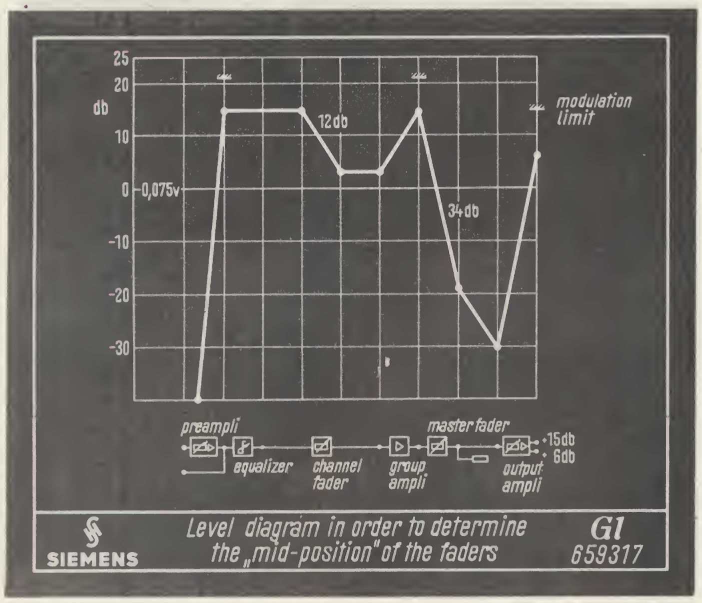


Fig. 3

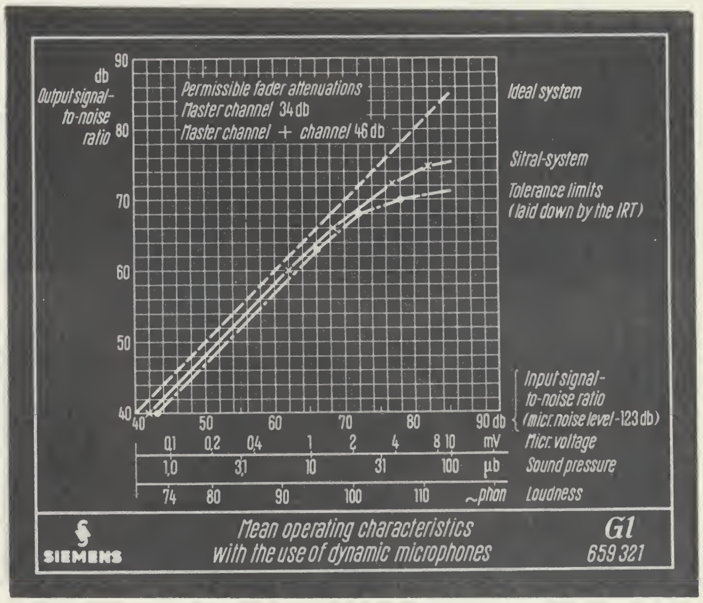
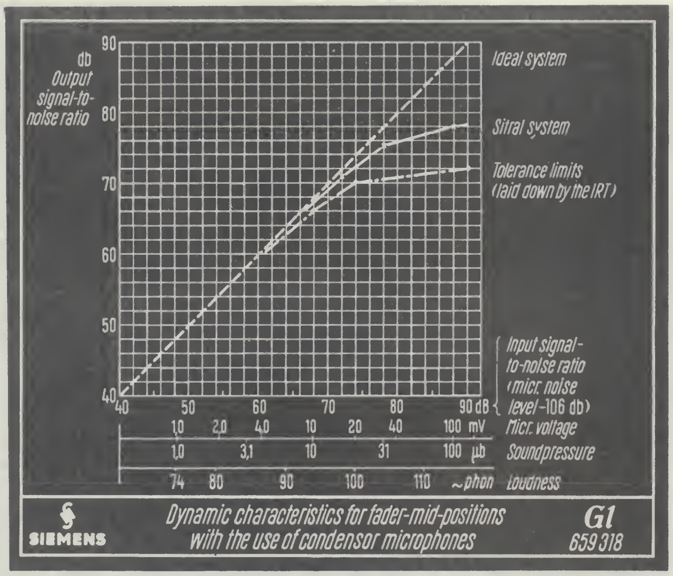


Fig. 4



Mean operating characteristics
with the use of condenser microphones

The slide has the same text as shown above. However, in conformity with Fig. 4 the lower text is preferable. It ought to be used when printing the paper.

Fig. 5

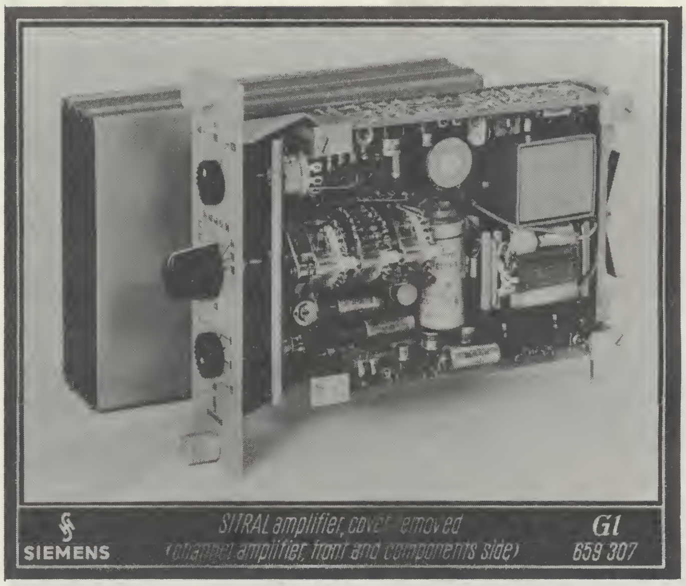


Fig. 6a

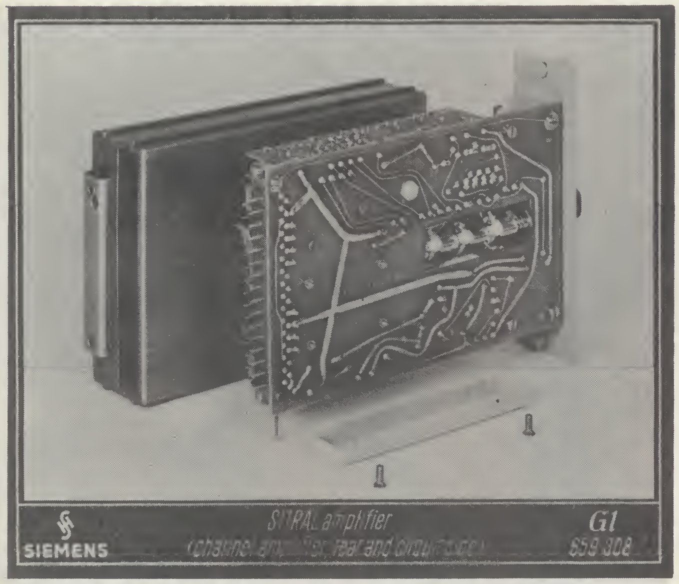


Fig.6b

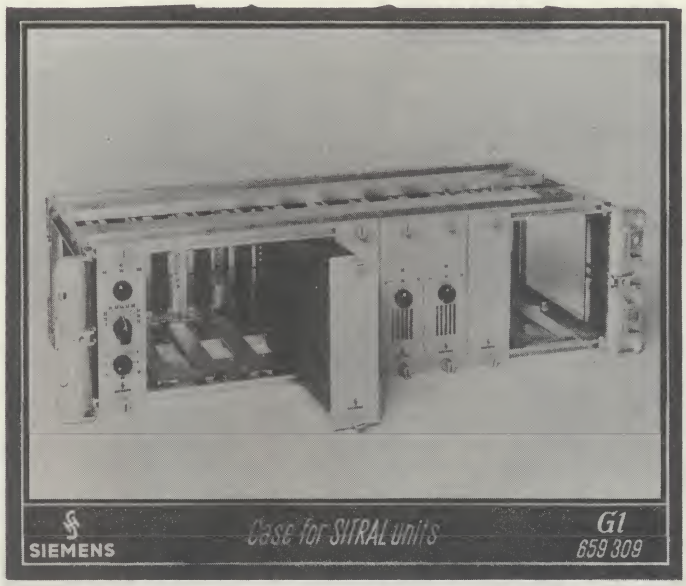


Fig. 7

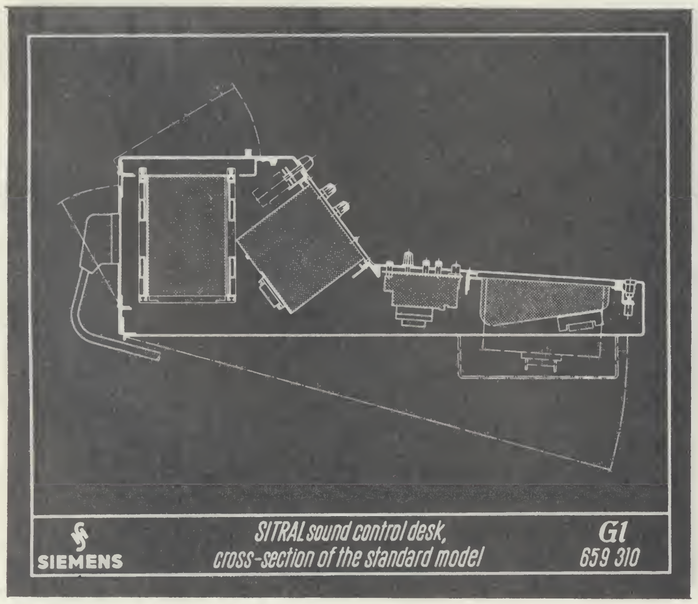


Fig. 8

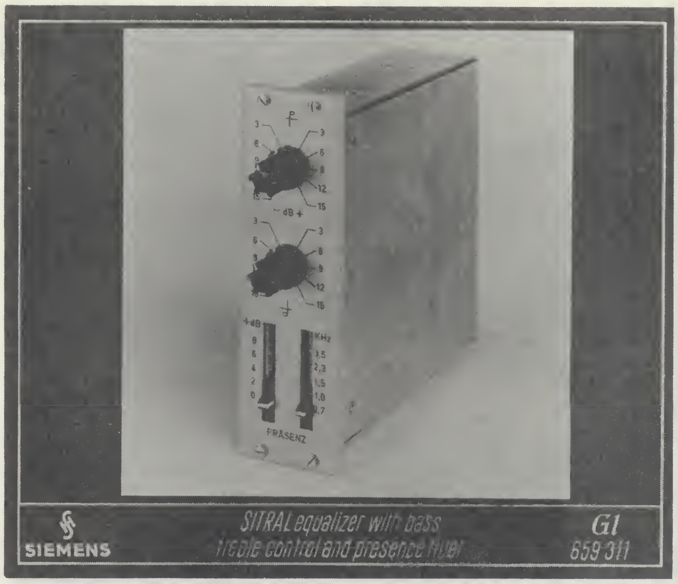


Fig. 9

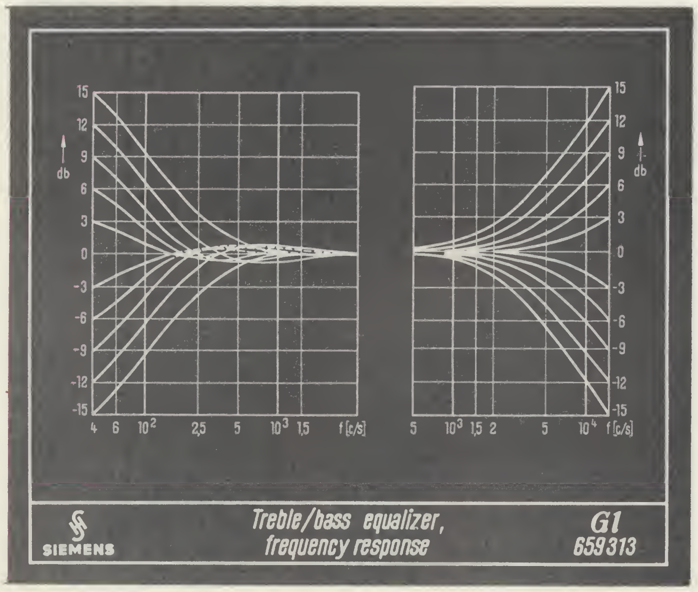


Fig.10a

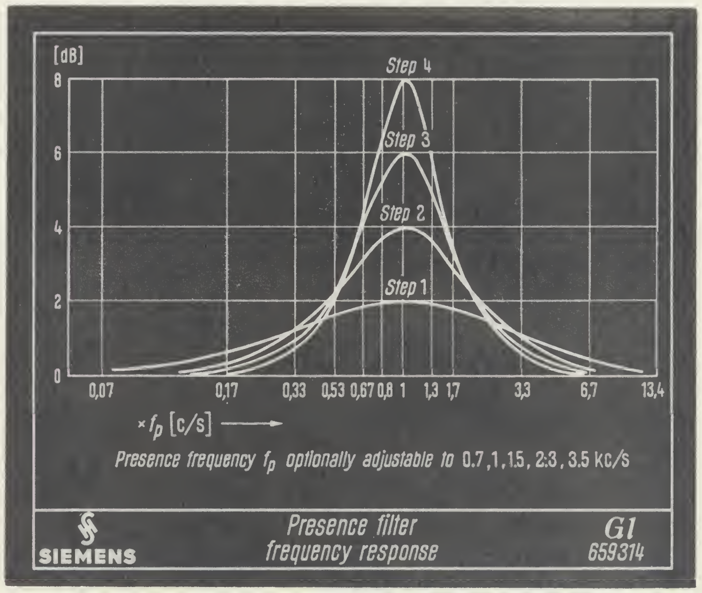
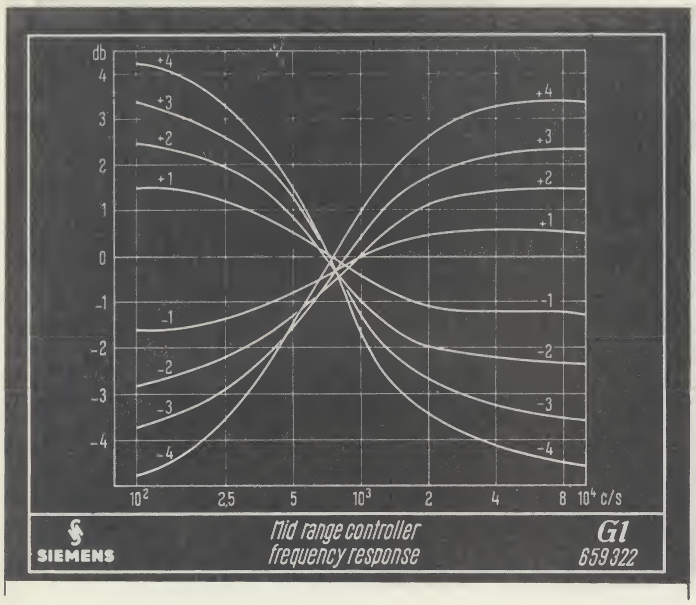
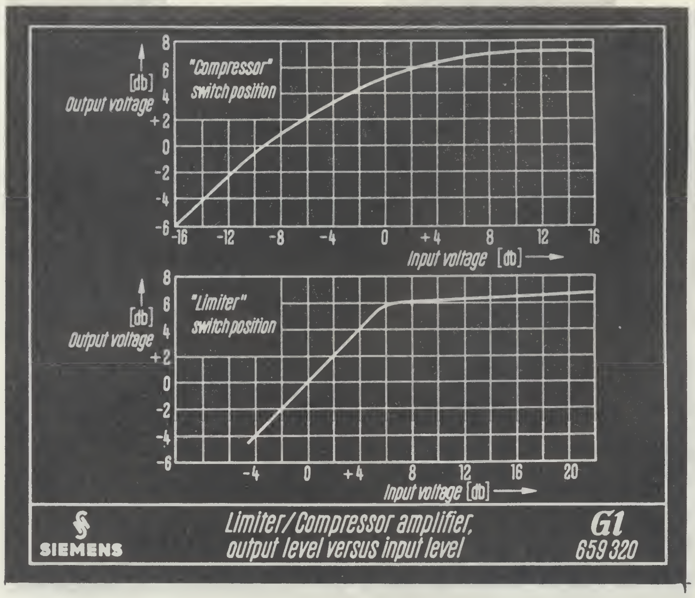


Fig. 10b



Mid range controller frequency response

Fig.loc



Limiter/Compressor amplifier, output level versus input level

Fig.11

6+2/2-channel sound control desk with SITRAL units

Fig. 12

12/2-channel sound control desk in SITRAL technique

Fig.13

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LIGHTWEIGHT SYNCHRONOUS STEREO RECORDING SYSTEM

BY

R.R. EPSTEIN, L. O'DONNELL and L. GREEN

NATIONAL FILM BOARD OF CANADA,
MONTREAL, QUE.

PRESENTED AT THE *SMPTE*
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LIGHTWEIGHT SYNCHRONOUS STEREO RECORDING SYSTEM

by R.R. Epstein, L. O'Donnell, L. Green, National Film Board, Montreal.

November 2nd, 1965.

For the Universal and International Exhibition of 1967, known as EXPO 67, the National Film Board of Canada will produce one of the theme shows entitled "LABYRINTH". It is planned to make extensive use of stereophonic sound as an aid to the multi-screen presentations in the motion picture theatres of this pavillion.

A production unit is already covering Canada and many other countries to collect suitable material for these film presentations. For the location recordings, a staff sound man has been assigned to the production crew. It was decided to use two-track stereophonic recording equipment with the multiple camera system,[1,2] which will be described in another session of this conference. Conventional multi-track equipment is too bulky and heavy, and requires external power for the capstan drive and torque motors. This presents a problem on remote locations. Also, most professional equipment does not lend itself to one-man operation.

Over the last five years the Sound Division of the National Film Board has used Nagra-Kudelski $\frac{1}{4}$" recorders, powered by flashlight batteries, for synchronous location sound work. This equipment has proved extremely reliable under the most difficult conditions and satisfies the requirement for a small, lightweight, synchronous recording system. Since a stereo version of the Nagra could not be supplied by the manufacturers, it was decided to adapt a standard Nagra 111 BH recorder for this purpose.

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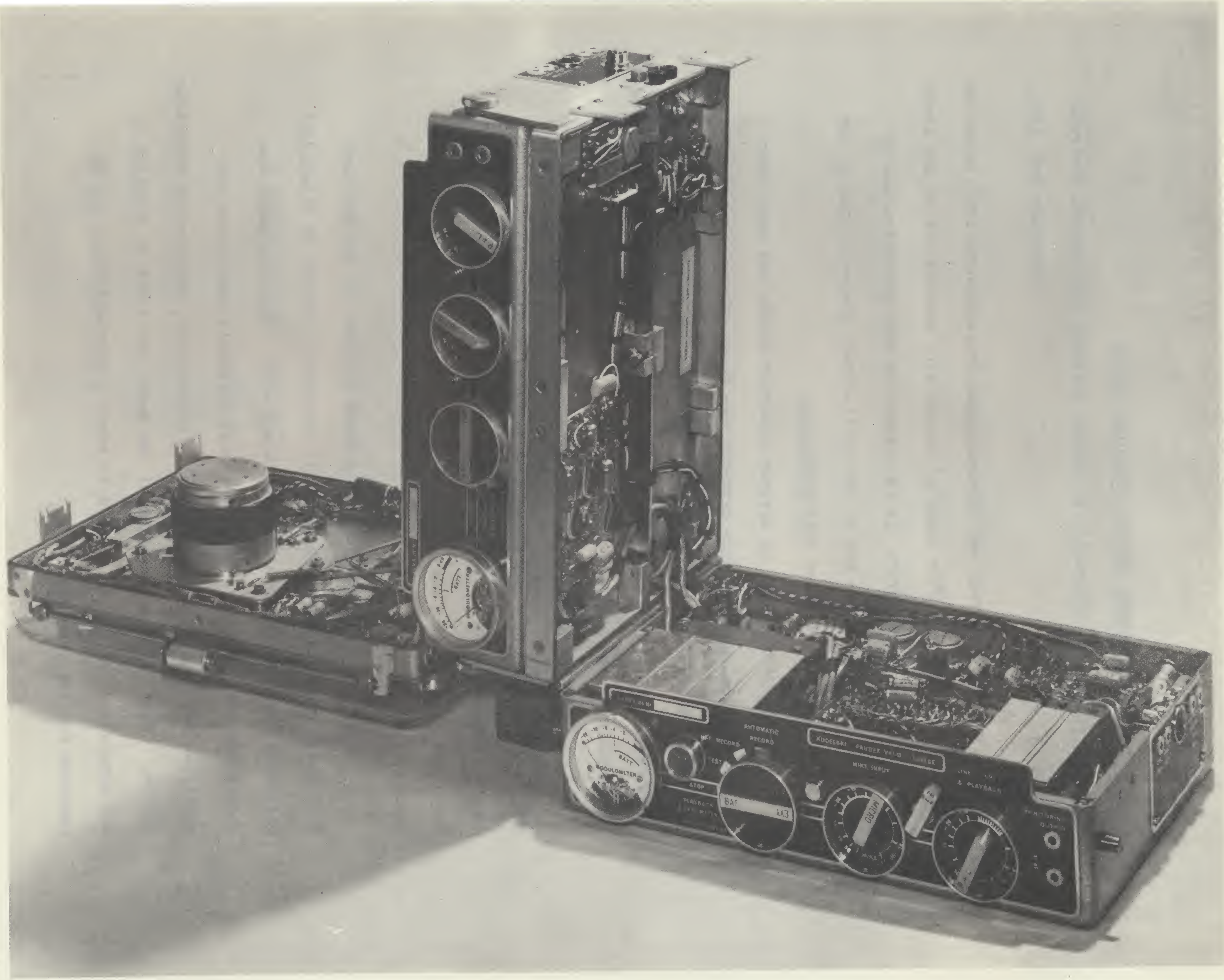

Fig. 1 Stereo Nagra opened for service.

LOCATION RECORDER

To accomplish the stereo modification the additional channel electronics were obtained from the Swiss manufacturers in the form of a second pre-wired Nagra. In this fashion the recording and playback amplifiers, metering circuits, mode switching and monitoring facilities were supplied. As we were only concerned with the audio system for the second channel, it was delivered without the drive motor, speed control equipment, recording and playback heads. The second unit was secured beneath the "complete" recorder by means of a hinged mounting plate which replaces the original lower tape deck. All components normally mounted on the underside of the tape deck were relocated on the mounting plate. In this manner, both units were securely combined, but access to both battery compartments and the internal circuitry is possible in the usual way. (Fig.1). As the existing mounting threads on the Nagras were used, any surgery of the equipment cases could be avoided. The erase head assembly of the top unit was left unaltered. The regular single track record and playback head assemblies were removed and substituted with two-track stereo heads. These were manufactured by Wolfgang Bogen Company, Berlin, to the same head inductances as the standard Nagra. Precise headmounts were designed for installation of these heads on the Nagra tape deck. (Fig.2).

The regular monophonic full-track Nagra recorders employ the Neo-Pilot method of control signal in which a biased push-pull recording is applied to the center of the $\frac{1}{4}$" tape. In the case of full-track recording, the Pilot frequency does not interfere with the program channel. [3] However, in a two-track or half-track recording, the long flux lines of the 60 c/s control track frequency would stray into each program channel. This reduces the S/N ratio. A system to overcome this difficulty in multi-track recording was already available from Telefunken of Germany. [4] The Telefunken system utilized the septum between the two program tracks to record a 10 kc carrier which is frequency modulated by the 60 c/s synchronizing signal. (Fig.3).

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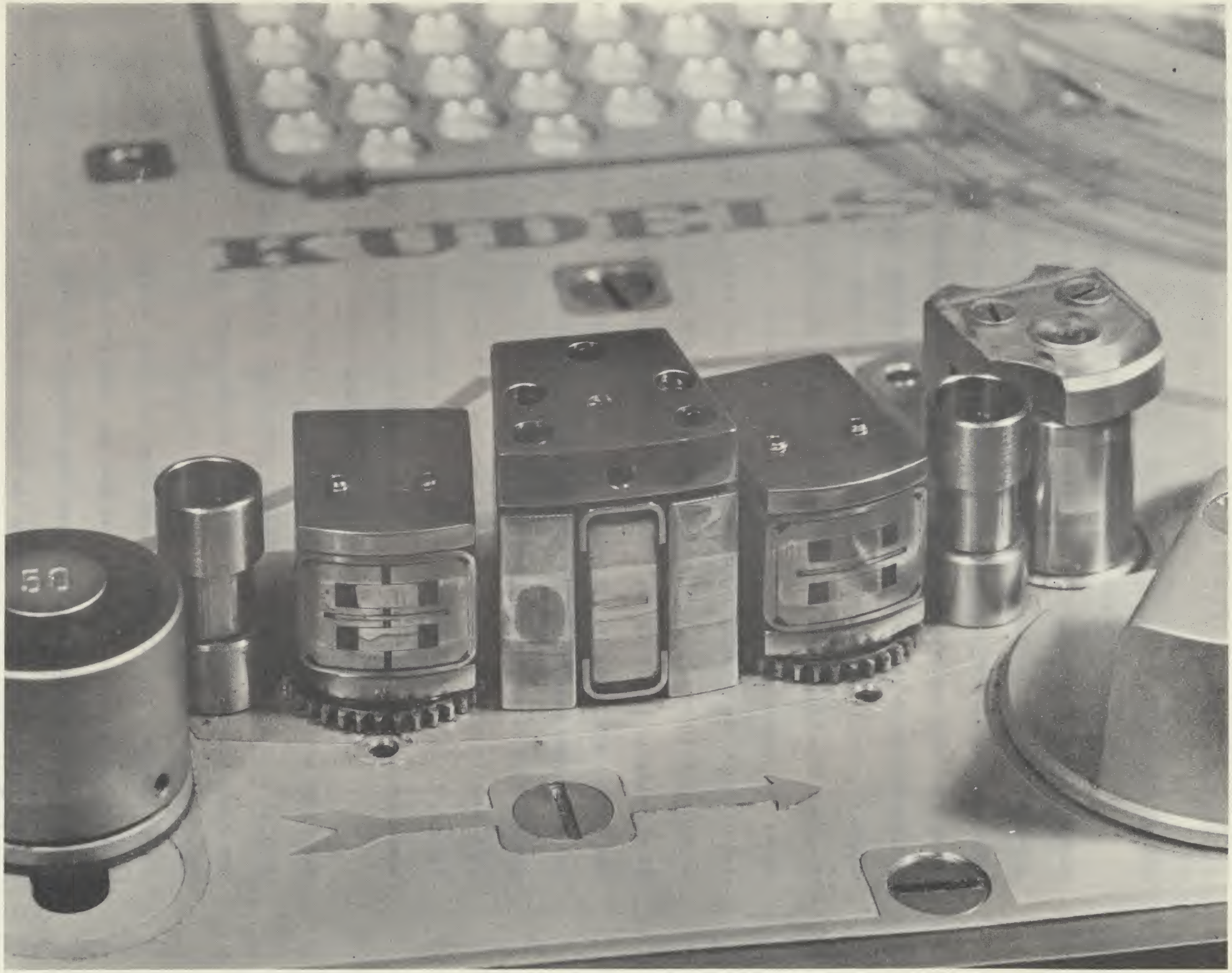

Fig. 2 Close-up of stereo heads and F.M. pilot head mounted on Nagra deck.

Incorporated in the Nagra package were a carrier oscillator for the recording of this FM track and a demodulator for playback of the synchronous recordings. For recording it is only necessary to apply the 60 c/s control signal to the Pilot-Tone input of the equipment in the usual manner. For playback the demodulated output is available on the Pilot-Tone receptacle. To provide synchronous playback at any location a Nagra synchronizer was modified for battery operation and powered from the regulated supply of the top unit. The additional drain for this synchronizer is only in the order of 12 ma. The reference frequency input on the synchronizer was designed to have the same characteristics as a Nagra Pilot-Tone input. Therefore, it can accept a standard signal from any Pilot-Tone generator. A Nagra quartz generator, also powered from the regulated supply, provides an accurate 60 c/s reference frequency to match the speed control of the multi-camera rig.

The internal loudspeakers were removed from the recorders, as it was not possible to incorporate them in the double Nagra package. The speaker connections were brought out to connectors at the side of the recorder, and two speakers were mounted in wooden boxes which can be easily set-up on location. (Fig.4).

To minimize circuit changes the use of the upper and lower function switches was retained. Therefore, it was necessary to provide that the tape transport would not run until both switches were selecting the same function. This was achieved by rearrangement of the DC switching circuits. The ground return of the motor control amplifiers passes in series through corresponding switch contacts in the upper and lower banks.

As the Telefunken FM pilot head required accurate azimuth and position adjustments, a suitable headmount was designed. Additional shielding in front of the FM head proved to be necessary to eliminate the induction of the carrier frequency into the playback heads.

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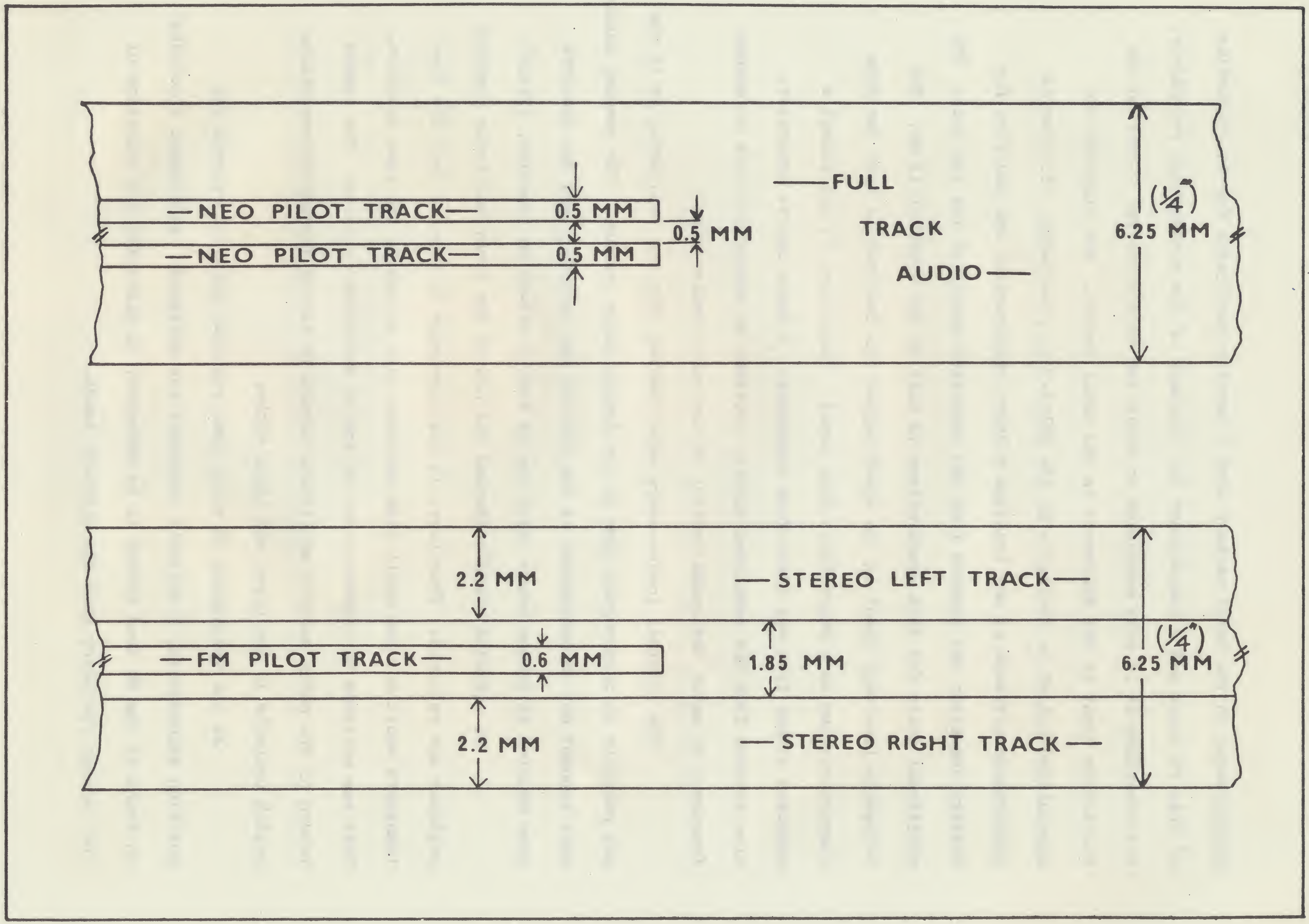

NEO PILOT TRACK
0.5 MM
0.5 MM
NEO PILOT TRACK
0.5 MM
FULL
TRACK
AUDIO
(1/4")
6.25 MM
2.2 MM
STEREO LEFT TRACK
FM PILOT TRACK
0.6 MM
1.85 MM
(1/4")
6.25 MM
2.2 MM
STEREO RIGHT TRACK

Sufficient HF bias to drive both sections of the stereo record head is available from the one original bias oscillator. Connecting the bias coupling capacitors to the common bias drive point does not cause any deterioration in the cross talk between channels. It was necessary to install the lower channel bias trap on the head coupling circuit board located on the underside of the tape deck. This board was almost completely rebuilt to accommodate the additional parts.

In order to fit the electronics for the FM system in the space available three circuit boards were constructed. The FM oscillator board was installed on the underside of the mounting plate in the lower compartment. The FM playback amplifier and demodulator boards were housed in two aluminium boxes and mounted in place of the motor speed control modules.

Fig. 5 shows the circuit of the FM carrier oscillator.[4] The transistors T1, T2 and T3 deliver a saw-tooth voltage. Capacitor C is charged with constant current through T1. The rate of charge can be adjusted by the variable resistor in the base circuit of T1. There is a linear relationship between the frequency of the saw-tooth oscillator and the charging current of C, which is controlled by T1. Therefore, the frequency of the oscillator is proportional to the instantaneous amplitude of the input signal. The saw-tooth output is then limited by T4 and T5 to generate a square-wave which drives the FM pilot head through a blocking capacitor and a resistor. As this is an FM recording system no HF bias is necessary. The demodulator schematic is illustrated in Fig. 6.[4] The 10 kc signal from the FM playback amplifier is applied to a diode limiter and then amplified further to operate a Schmitt trigger. This converts the carrier to a square-wave of equal amplitude at all input frequencies. Then through a phase-splitter stage and a class B final amplifier the square-wave signal is coupled to a full-wave bridge rectifier. The demodulator, at this point, works on the principle of the direct reading frequency meter and separates the modulation from the carrier.

Continued 5

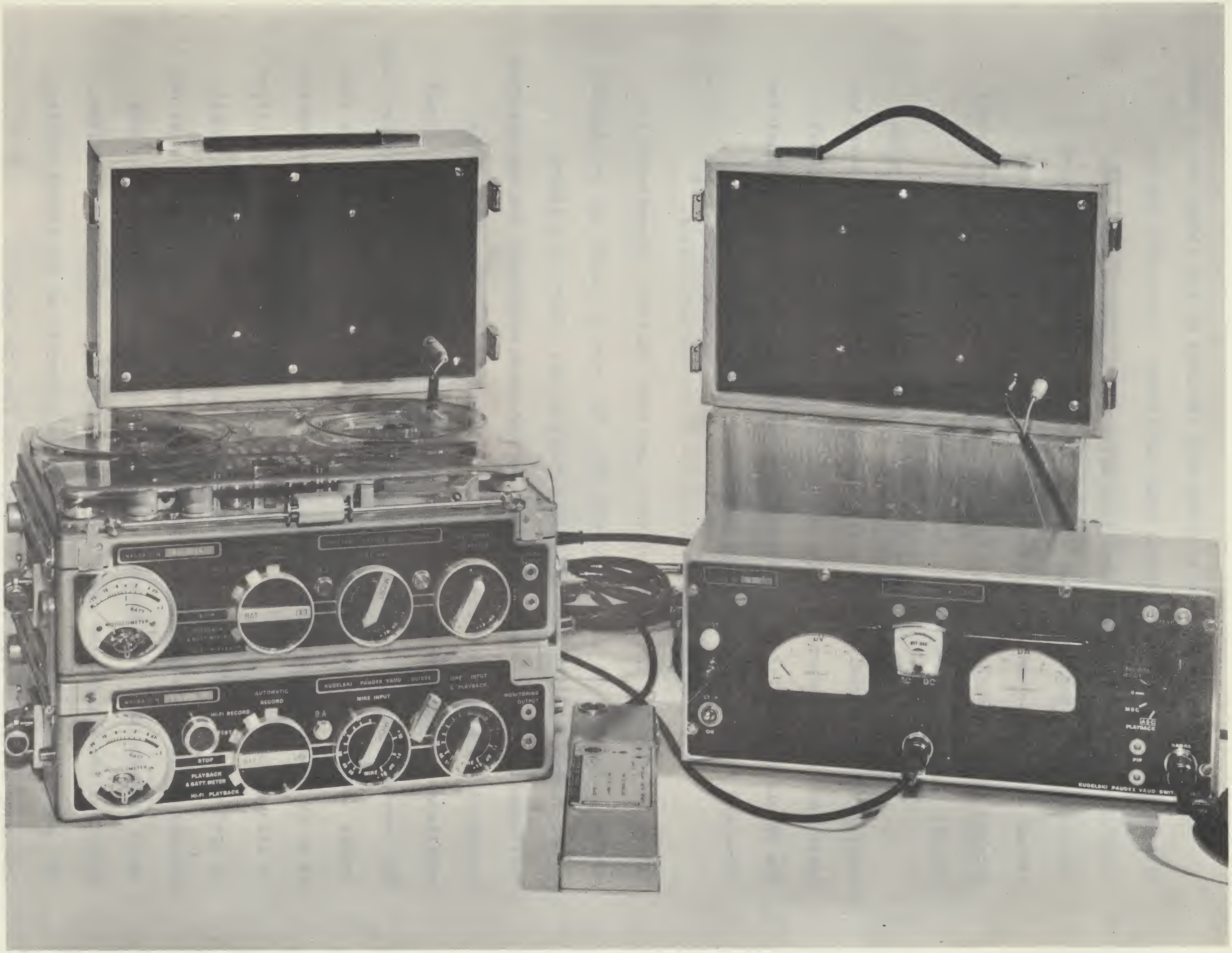


Fig. 4 Stereo Nagra with synchronizer, quartz 60 cps generator and playback speakers.

The adjustable ferrite cores which normally function as the speed control adjustments for the Nagra were put into use here as two of the inductive elements of the low pass filter on the output of the FM demodulator. A third inductor was added to complete the filter and the output padded down to a suitable level for the synchronizer.

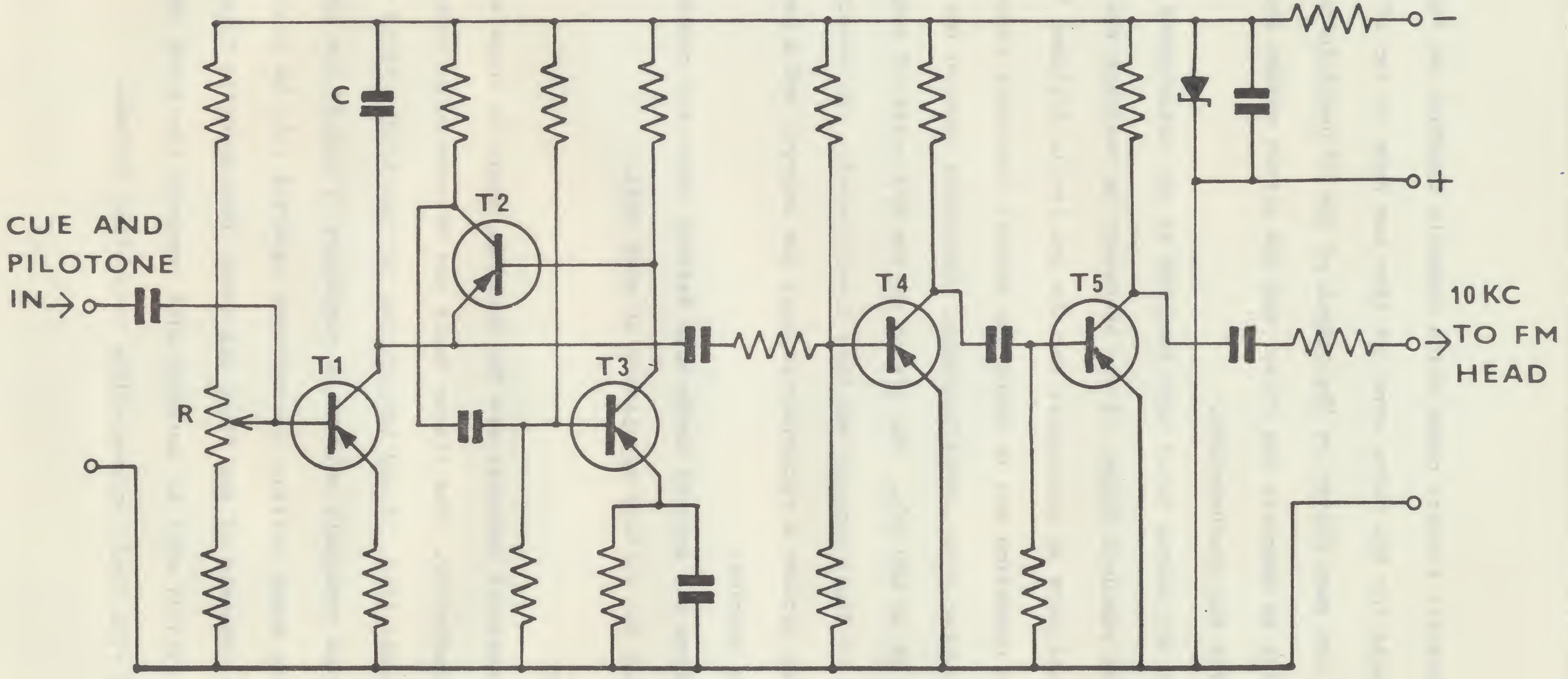
Although the stereo heads were supplied at the inductances as specified for the standard Nagra, it was necessary to readjust the recording and playback equalizers to compensate for the new levels dictated by the change to stereo operation and to bring the overall frequency response in line with the regular Nagra specifications. Crosstalk between the two stereo channels was 60 db at 400 c/s. The effect on the S/N ratio of both stereo channels by the FM Pilot carrier was also investigated. No measurable change could be observed between a recording without the carrier and a recording with the modulated FM carrier.

To equalize the battery drain both battery packs are connected in parallel and feed the voltage regulators of each unit.

STUDIO EQUIPMENT

At the Montreal headquarters the location sound is transferred for editing and re-recording. The stereo tapes are reproduced on Ampex 354 two-track equipment (Fig. 7) and transferred to 35mm triple-track magnetic film. To minimize crosstalk only track positions #1 and #3 are used at this stage. Following sound editing the selected material will be distributed to the appropriate channels of the final six track 35mm magnetic film. Each theatre at the exhibit will be equipped with separate six-track reproducers and interlocked with their corresponding projection systems.

Continued 6



10 KC. FM OSCILLATOR

The Ampex 354 is a standard two-track stereo recorder. For the transfer of the location tapes a matching Telefunken FM Pilot-head and the complete Telefunken carrier electronics were installed on this equipment. During playback the output of the demodulator connects to a Magna-Tech playback synchronizer which controls the speed of the capstan motor in the usual manner.

ADDITIONAL APPLICATIONS OF FM PILOT SYSTEM

In addition to the transfer operation, this same two-track stereo equipment can be used for synchronous recording. Furthermore, as the FM carrier will accommodate modulating frequencies up to 3,000 c/s., this feature can be utilized in multi-camera operations by equipping each camera with a transistorized oscillator operating at an identifying frequency. One system where this device is used records the oscillator tone on the FM pilot track whenever it's own particular camera is switched on.[6] For other occasions it has proved more suitable to record a beep at the camera start, or to use one of the more conventional slate-light arrangements. At the same time the instructions between the film director and his camera crew can be recorded as an editing aid. The interphone is connected to the FM system input through a 200 c/s.to 3,000 c/s. band-pass filter to prevent interference with the Pilot frequency.

These applications were illustrated in a recent production of the National Film Board. A classical piano concerto was filmed with four 16mm cameras and recorded on both program channels. Identification signals for each camera and the interphone conversation were added to the 60 c/s pilot signal. During the transfer to 16mm magnetic film an auxilliary copy served as an editing guide. This guide copy reproduced the camera signals and the interphone instructions.

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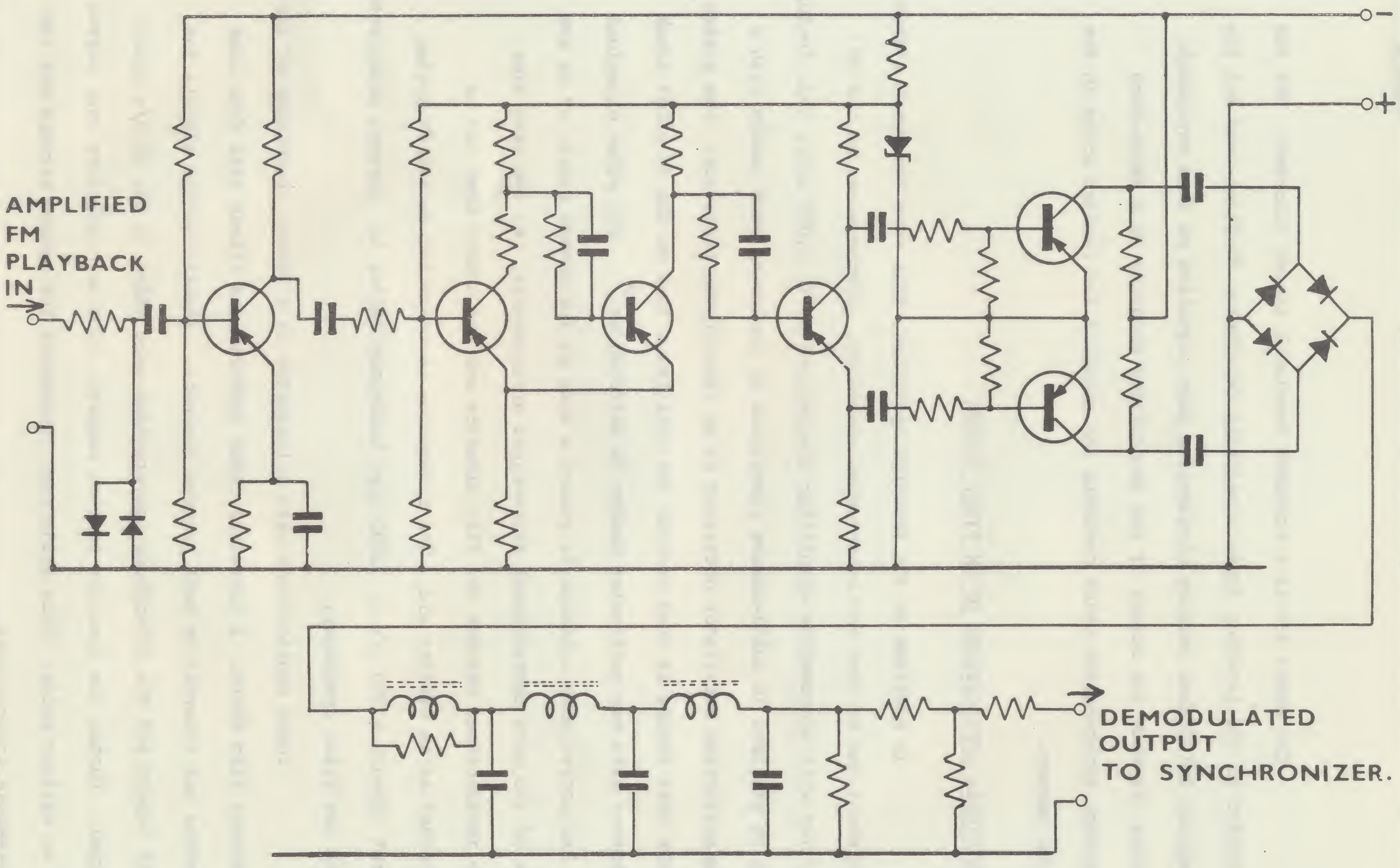

10 KC. FM DEMODULATOR.

The same equipment can be used for monophonic recording or playback. The Headmount for the FM Pilot head was designed to permit easy and accurate change-over from the Telefunken FM head to Neo-Pilot or other control track heads. (Fig. 8). The Ampex stereo head assembly can be replaced with a matching full track headmount.

The authors wish to express their appreciation for the assistance and helpful suggestions received from Nagra-Kudelski, Telefunken A.G. and W. Bogen G.m.b.H. in connection with this project.

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Fig. 7 Ampex 354 stereo equipment with Telefunken F.M. pilot synch system.

LIST OF ILLUSTRATIONS

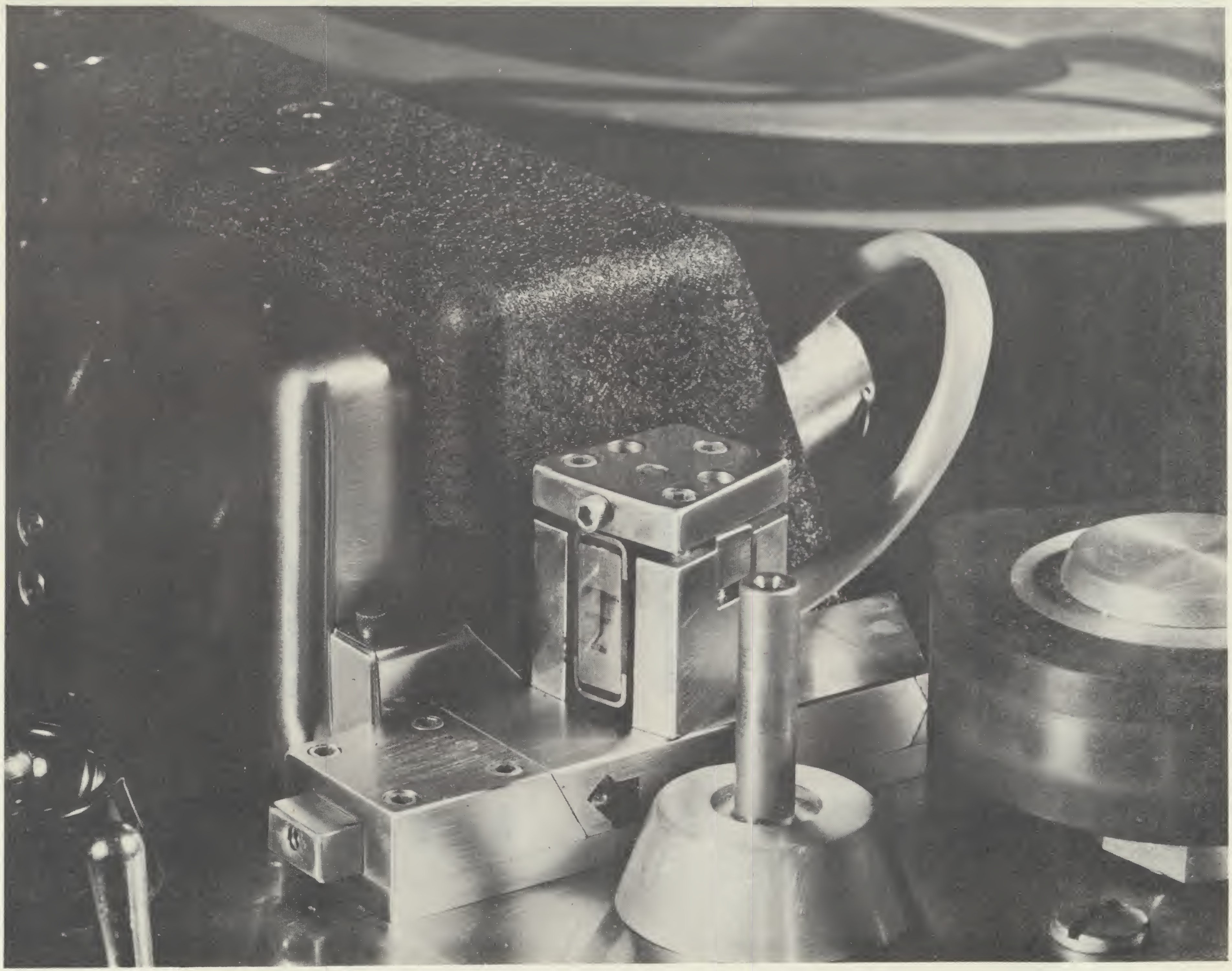

Fig. 8 Close-up of F.M. pilot head in snap-in mounting on Ampex deck.